CISCO CONFIDENTIAL

Patent Idea Details for Idea #74421

General Information

Title: User controlled audio quality for Voice-over-IP telephony systems

ID: 74421

Patent No.: ---, ---

URL: [Application No. ---]

Inventors: David Oran (oran)
More details on these inventors listed below.

Date Entered:

Date Modified:

Date Filed: ---

Date Issued: ---

Background: Classic PSTN telephony offers exactly one level of service, based on the use a single audio quality regime (64kbps PCM audio with 300-3400hz passband) and control regime which deterministically denies service rather than degrading service when failures or congestion occurs.

Summary: Voice-over-IP systems can be built to mimic this service regime, but are inherently much more flexible, as the protocols and algorithms can be made adaptive in various ways. Example adaptations include:

- either deny service or degrade when resources cannot be reserved
- switch codecs mid-call if bandwidth becomes more or less scarce
- add or remove FEC (forward error correction) as error rates change
- change packetization interval to either limit bandwidth or packet-per-second processing load

To date, these adaptations have either been statically provisioned by the network designer, or invoked under control of the network operator. This invention proposes a set of methods by which the user can control the tradeoffs directly. In adjusting these parameters, the control method has to take the effects of network congestion into account. While it is tempting to just "crank up the gain" by injecting more packets to attempt to improve quality, this can both be damaging to the network and counter-productive to improving audio quality. Therefore all the techniques used

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in the invention take account of congesion feedback (through jitter/loss measurements conveyed in control protocols such as RTCP) and either isolate the user from congestion (by modulating RSVP reservations) or reacting to congestion by modulating packetization, FEC, and codec.

Advantages: Conceptually, I propose a "knob" which the user can control both before *and during* a voice-over-IP telephone call to control the adaptation algorithms cited above. The knob would have two extreme settings, and a number of intermeidate settings. Proposed settings would include:

extreme one way:
  use best effort traffic, cheapest coder, no FEC, longish packetization

extreme other way:
  don't complete call unless resources locked down, short packetization,

  high quality coder
intermediate 1:
  try to get reservation but accept best effort while trying, switch to lower-bandwidth coder and try for reservation at lower bandwidth
intermediate 2:
  same as 1 except add FEC

different embodiments could have different adaptations for different settings and be as elaborate as the implementor wished. Among the preferred embodiments would be one that used linear regression based on current measurements to find the optimal adaptation points for each of the adaptation mechanisms, where the knob essentially controls the weighting coefficients (note: prior art on using linear regressions to control voip adaptation - c.f. Shuster patent). The measurement includes
estimation of network congestion (as noted in the summary above)
and ensures
that the adaptation envelope stays within acceptable congestion bounds. Such bounds are usually established through either the absolute packet loss
rate or the first derivative thereof (e.g. if the loss rate is increasing

adapt by cutting down on total bandwidth consumed)

The embodiment of the conceptual "knob" would differ depending on the instrument the user was employing for VoIP. Specifically:

IP Phone: a physical knob or screen control on the instrument
PC softphone: a graphical slider or similar iconic representation
POTS phone: DTMF signaling to the gateway (perhaps after a hook flash to indicate a mode switch), similar to the controls used by conferencing facilities to control mute, etc.

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In a service-provider environment where the provider can offer differentiated services with different billing regimes for differing quality, the user can use the knob to control how much he is being charged for a call either before *or during* the call.

Cisco Use:

Method of Detecting Use By Other Companies:

Previous Public Use: none

First Written Record Date:

First Written Record URLs: ---

Supporting Docs URLs: ---

Notes: ---

Inventor Details: David Oran (oran) — See also Cisco Directory

Location:

Phone:

Division: SP Systems & Solutions Eng

Manager: Systems & Solutions Eng

Review Information

Review Progress:

Group Name: SPLOB-TISU

Users:
caryfitz, clayn, dee, gronski, lsantora, oran*, tthio

(Review Done)

1 node, 100% complete.

**User oran is a designated reviewer for SPLOB-TISU. (Other members of SPLOB-TISU may*

This idea has been approved. The status of this application will not be updated in CPOL until after it has been filed with the US Patent Office. All post-approval information is stored in an off-line database. If you have questions, please use Feedback to contact the patent attorneys.

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also review this Idea.)

Reviewers' Comments: REASON FOR APPROVAL:
Will allow user to better control voice quality for VoIP leading to a better service offering by Cisco customers.

OTHER NON-CISCO USERS OF THIS INVENTION:
Other VoIP equipment vendors

DETECTION OF NON-CISCO USE:

Please add comments in this area

Designated Reviewer:

Manager List:

Admin Flags:

Needs edit	☐
Done approval	☑
Rejected	☐
Publicly viewable	☐
Active	☑
Discarded	☐

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